What Is Claimed Is:

1. A method for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising the steps of:

determining whether a called party's audio device is able to support at least one voice compression algorithm supported by a calling party's audio device; and

exchanging voice signals between said called party's audio device and said calling party's audio device via a data network, if said called party's audio device is able to support said at least one voice compression algorithm.

2. The method of claim 1, wherein said determining step is accomplished by exchanging messages between said called party's audio device and said calling party's audio device via the circuit switched network.

3. The method of claim 2, wherein said messages are modified circuit control signaling messages.

4. The method of claim 1, wherein said exchanging step comprises the steps of:

compressing said voice signals using said at least one voice compression algorithm at one of said called party's audio device and said calling party's audio device;

sending said compressed voice signals to one other of said called party's audio device and said calling party's audio device via said data network; and

decompressing said compressed voice signals using said at least one voice compression algorithm at said one other of said called party's audio device and said calling party's audio device.

5. The method of claim 1, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

6. The method of claim 1, wherein said data network is an Internet protocol (IP) network.

7. A method for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising the steps of:

sending a call request message from a calling party's audio device to a first line/network interface switch, said call request message identifying at least one voice compression algorithm supported by said calling party's audio device;

receiving said call request message at said first line/network interface switch;

sending said call request message from said first line/network interface switch to a second line/network interface switch via the circuit switched network;

receiving said call request message at said second line/network interface switch;

sending said call request message from said second line/network interface switch to a called party's audio device;

receiving said call request message at said called party's audio device;

sending a response message from said called party's audio device to said one of said first line/network interface switch and said calling party's audio device via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

exchanging voice signals between said called party's audio device and said calling party's audio device via a data network, if said called party's audio device is able to support said at least one voice compression algorithm.

8. The method of claim 7, wherein said first line/network interface switch and said second line/network interface switch are mobile switching centers (MSCs).

9. The method of claim 7, wherein the step of sending said call request message from said first line/network interface switch to a second line/network interface switch via the circuit switched network includes the step of modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm; and

wherein the step of sending a response message from said called party's audio device to said calling party's audio device via the circuit switched network includes the step of modifying at least one circuit control signaling message to include response information.

10. The method of claim 7, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

11. The method of claim 7, wherein said data network is an Internet protocol (IP) network.

12. A method for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising the steps of:

sending a call request message from a calling party's audio device to a first line/network interface switch, said call request message identifying at least one voice compression algorithm supported by said calling party's audio device;

receiving said call request message at said first line/network interface switch;

sending said call request message from said first line/network interface switch to a second line/network interface switch via the circuit switched network;

receiving said call request message at said second line/network interface switch, said second line/network interface switch being associated with a called party's audio device;

sending a response message from said second line/network interface switch to one of said calling party's audio device and said first line/network interface switch via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

exchanging voice signals between said called party's audio device and said calling party's audio device via a data network, if said called party's audio device is able to support said at least one voice compression algorithm.

13. The method of claim 12, wherein said first line/network interface switch and said second line/network interface switch are mobile switching centers (MSCs).

14. The method of claim 12, wherein the step of sending said call request message from said first line/network interface switch to a second line/network interface switch via the circuit switched network includes the step of modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm; and

wherein the step of sending a response message from said called party's audio device to said calling party's audio device via the circuit switched network includes the step of modifying at least one circuit control signaling message to include response information.

15. The method of claim 12, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

16. The method of claim 12, wherein said data network is an Internet protocol (IP) network.

17. A method for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising the steps of:

receiving a call request message at a first line/network interface switch;

modifying said call request message to identify at least one voice compression algorithm supported by said calling party's audio device;

sending said modified call request message from said first line/network interface switch to a second line/network interface switch via the circuit switched network;

receiving said modified call request message at said second line/network interface switch;

sending said modified call request message from said second line/network interface switch to a called party's audio device;

receiving said modified call request message at said called party's audio device;

sending a response message from said called party's audio device to one of said calling party's audio device and said first line/network interface switch via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

exchanging voice signals between said called party's audio device and said calling party's audio device via a data network, if said called party's audio device is able to support said at least one voice compression algorithm.

18. The method of claim 17, wherein said first line/network interface switch and said second line/network interface switch are mobile switching centers (MSCs).

19. The method of claim 17, wherein the modifying step is accomplished by modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm; and

wherein the step of sending a response message from said called party's audio device to said calling party's audio device via the circuit switched network includes the step of modifying at least one circuit control signaling message to include response information.

20. The method of claim 17, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

21. The method of claim 17, wherein said data network is an Internet protocol (IP) network.

22. A method for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising the steps of:

sending a call request message from a calling party's audio device to a first line/network interface switch;

receiving said call request message at said first line/network interface switch;

modifying said call request message to identify at least one voice compression algorithm supported by said calling party's audio device;

sending said modified call request message from said first line/network interface switch to a second line/network interface switch via the circuit switched network;

receiving said call request message at said second line/network interface switch, said second line/network interface switch being associated with a called party's audio device;

sending a response message from said second line/network interface switch to one of said calling party's audio device and said first line/network interface switch via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

exchanging voice signals between said called party's audio device and said calling party's audio device via a data network, if said called party's audio device is able to support said at least one voice compression algorithm.

23. The method of claim 22, wherein said first line/network interface switch and said second line/network interface switch are mobile switching centers (MSCs).

24. The method of claim 22, wherein the modifying step is accomplished by modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm; and

wherein the step of sending a response message from said called party's audio device to said calling party's audio device via the circuit switched network includes the step of modifying at least one circuit control signaling message to include response information.

25. The method of claim 22, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

26. The method of claim 22, wherein said data network is an Internet protocol (IP) network.

27. A method for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising the steps of:

receiving a call request message from a calling party's audio device, said call request message identifying at least one voice compression algorithm supported by said calling party's audio device;

sending said call request message to one of a called party's audio device and a line/network interface switch associated with said called party's audio device via the circuit switched network;

receiving a response via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

enabling said calling party's audio device and said called party's audio device to exchange voice information via a data network if said called party's audio device is able to support said at least one voice compression algorithm.

28. The method of claim 27, wherein said sending step includes the step of modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm.

29. The method of claim 27, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

30. The method of claim 27, wherein said data network is an Internet protocol (IP) network.

31. A method for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising the steps of:

receiving a call request message from a calling party's audio device;

modifying said call request message to identify at least one voice compression algorithm supported by said calling party's audio device;

sending said modified call request message to one of a called party's audio device and a line/network interface switch associated with said called party's audio device via the circuit switched network;

receiving a response message via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

enabling said calling party's audio device and said called party's audio device to exchange voice information via a data network if said called party's audio device is able to support said at least one voice compression algorithm.

32. The method of claim 31, wherein said modifying step is accomplished by modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm.

33. The method of claim 31, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

34. The method of claim 31, wherein said data network is an Internet protocol (IP) network.

35. A method for diverting an Integrated Services Digital Network User Part (ISUP) network talkpath to a data network talkpath, comprising the steps of:

determining, using an ISUP signaling path, whether a called party's telephone is adapted to exchange voice signals via a same data network to which a calling party's telephone is adapted to exchange voice signals, said ISUP signaling path being established during a process of establishing the ISUP network talkpath;

establishing the data network talkpath using resources associated with said same data network if said called party's telephone is adapted to exchange voice signals via said same data network; and

exchanging voice signals between said called party's telephone and said calling party's telephone using the data network talkpath.

36. A computer-readable medium whose contents cause a computer system to improve a quality of voice signals exchanged via a circuit switched network between audio devices, by performing the steps of:

receiving a call request message from a calling party's audio device, said call request message identifying at least one voice compression algorithm supported by said calling party's audio device;

sending said call request message to one of a called party's audio device and a line/network interface switch associated with said called party's audio device via the circuit switched network;

receiving a response via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

enabling said calling party's audio device and said called party's audio device to exchange voice information via a data network if said called party's audio device is able to support said at least one voice compression algorithm.

37. The computer-readable medium of claim 36, wherein said sending step includes the step of modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm.

38. The computer-readable medium of claim 36, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

39. The computer-readable medium of claim 36, wherein said data network is an Internet protocol (IP) network.

40. A computer-readable medium whose contents cause a computer system to improve a quality of voice signals exchanged via a circuit switched network between audio devices, by performing the steps of:

receiving a call request message from a calling party's audio device;

modifying said call request message to identify at least one voice compression algorithm supported by said calling party's audio device;

sending said modified call request message to one of a called party's audio device and a line/network interface switch associated with said called party's audio device via the circuit switched network;

receiving a response message via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

enabling said calling party's audio device and said called party's audio device to exchange voice information via a data network if said called party's audio device is able to support said at least one voice compression algorithm.

41. The computer-readable medium of claim 40, wherein said modifying step is accomplished by modifying at least one circuit control signaling message to include information identifying said at least one voice compression algorithm.

42. The computer-readable medium of claim 40, wherein each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone.

43. The computer-readable medium of claim 40, wherein said data network is an Internet provider (IP) network.

44. An apparatus for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising:

a processor;

a computer readable memory segment adapted to be connected to said processor;

a diversion module included within said computer readable memory, said diversion module comprising computer program code segments which, when executed by said processor, implement the following steps:

receiving a call request message from a calling party's audio device, said call request message identifying at least one voice compression algorithm supported by said calling party's audio device;

sending said call request message to one of a called party's audio device and a line/network interface switch associated with said called party's audio device via the circuit switched network;

receiving a response via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

enabling said calling party's audio device and said called party's audio device to exchange voice information via a data network if said called party's audio device is able to support said at least one voice compression algorithm.

45. An apparatus for improving a quality of voice signals exchanged via a circuit switched network between audio devices, comprising:

a processor;

a computer readable memory segment adapted to be connected to said processor;

a diversion module included within said computer readable memory, said diversion module comprising computer program code segments which, when executed by said processor, implement the following steps:

receiving a call request message from a calling party's audio device;

modifying said call request message to identify at least one voice compression algorithm supported by said calling party's audio device;

sending said modified call request message to one of a called party's audio device and a line/network interface switch associated with said called party's audio device via the circuit switched network;

receiving a response message via the circuit switched network, said response message indicating whether said called party's audio device is able to support said at least one voice compression algorithm; and

enabling said calling party's audio device and said called party's audio device to exchange voice information via a data network if said called party's audio device is able to support said at least one voice compression algorithm.